The Devolution of RTT Provisions in M376 and ANPRM

Comments Submitted to Access Board

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Executive summary

Industry and Consumers worked together in the TEITAC and came up with consensus language for Real-time text that

1. Required Real-time text wherever there was Voice over IP.
2. Allowed industry to select whatever real-time text technology they wanted for each different system – but required that everyone using that system support the RTT format chosen for that system.
3. NEVER required Industry to add hardware to a product design in order to implement Real-time text;
   a. real-time text reception was only required if the product design specifications already call for inclusion of a display for other purposes, and
   b. RTT transmission was only required if the product design specifications already call for inclusion of the ability to generate text for other purposes.

This was an effective set of provisions that would have resulted in built-in real-time text coupled with the speech in basically all VoIP, while not requiring industry to add any hardware and minimal software. ***The result would have been almost universal access to VoIP by people who are deaf – but in addition – for people who are hard-of-hearing – including people who are aging and losing their hearing but unable to physically type effectively or unable to type at all (e.g. arthritis).***

Since then

Since the TEITAC meetings, the consensus language has been abandoned and the new provisions in M376 edited (accidentally or intentionally) to make them pretty much meaningless. The current language of M376 (prEN301549)

1. ***No longer requires interoperability***
   a. Each vendor can use a different format if they choose
      i. so an RTT call will only go through if both the caller and the called person (and all of the carriers and equipment in between) all happen to use the same standard. However, there is no requirement that they use the same one – nor that they convert their format to the format of neighboring systems or equipment. (like there is for voice communication)
   b. no reliable RTT from end to end will exist (like it does for voice)
   c. in an emergency – a person could not use RTT to call emergency services or a neighbor or whomever to help them or evacuate them or .... and have confidence (or even probability?) that the RTT call would go through (like they could if they could talk).
   d. This would put RTT in the same situation as IM where companies all choose their own standards – and you only can call someone who has an account on your IM type (unless you are a techie that knows how to crosswire the IM formats – or have accounts on all of them). It SHOULD have the same interoperability as the voice portion of the call it is attached to.
2. ***No longer require requires that all VoIP devices have RTT***
   a. Even where it would require NO additional hardware of any kind, RTT is not required (even though it only requires minimal software – and that is all available as open source software for adaption to their product)
3. ***No longer requires that RTT work on the same call as Voice***
   a. which is needed for captioned telephony
   b. and even more so by elders who cannot hear (well or at all) and who cannot type (old hands, arthritis etc).
4. ***and more (see below)***

In short, an opportunity to provide universal text conversation as part of every voice conversation will be lost – even though the cost would be de minimus for all new products once it is in place and in practice by a company. (Only costs are basically start up costs, learning, communication etc, when first incorporating into products. Like closed captioning, it will soon disappear into the core software and cost essentially nothing – which providing benefit to all.)

What does it take to fix the language?

Repairing the language would not be hard (editorially) – but the changes were deliberate according to discussions with industry, and they do not want the provisions changed back. As they are now, there is little that industry would have to do.

Two examples to illustrate

- **the M376 language no longer requires RTT on any ICT at all.**
    - With the current M376 language, ICT <u>without any RTT on it</u> will meet the RTT requirements if it is just POSSIBLE for an app from a third party to be created and added at some later time in the future, if that app would provide RTT completely separately from the call.
- **The RTT also does not have to work with voice.**
    - As long as you can make two calls (one voice and one text) simultaneously to another person – it would conform. Since some elders require both text and voice together --- this is unworkable. Imagine if every elder needed to make two calls to two different numbers with two different applications in order to make a voice phone call. One would never sell a single phone. But this is OK under current language.
    - Captioned Telephony also requires voice and text on the same call

There is concern that, for harmonization reasons, M376 language would be adopted in the US. These comments are therefore submitted to the Access Board (and copied to M376 and the FCC for their information).

Some of these problems have also crept into the ANPRM language – though less so. And these are highlighted below as well.

<u>Comments on M376 Provisions</u>

The current M376 - EN 301 549 language as released for vote:

1. **…essentially does not require RTT at all.**
   The provision that would require RTT, has a note that says it can be added later and still pass. Then there is a second note that says it can be completely separate. This was explained to mean it could be a separate program or a separate device like a computer.

   Thus this provision is of no real value as written. It is already possible to add some unrelated real-time text program to virtually every computer and phone today. So the provision doesn't add any requirement at all that isn't already true by default.

2. **…does not require that RTT and Voice be in any way associated with each other or be part of the same call.**
   a. As long as <u>some program</u> can do real-time text (or <u>some program can be added later</u> that can do Real-time text) -- and the two apps can be used at the same time -- then it is OK to ship a VoIP product with no text at all as part of it.   In fact, as written (and described in the drafting meeting as the intent) the RTT app does not have to even be on the same device as the phone call.
   b. There is no requirement that the voice and RTText be on the same call.

   Again this provision is of no practical effect since it is essentially technically impossible to fail this provision. One would have to create a VoIP program that would prevent the use of a completely separate text program at the same time. Hence, again, this provision requires nothing that isn't already true.

3. **…does not require any interoperability at all.**
   a. The "interoperability" provision says that you must use A or B or any other "published and available common specification for RTT exchange".
   b. Any telecom engineer will tell you that you cannot have two things interoperate that don't support the same format, (not even at their borders).
      i. That is similar to saying that you want everyone in a company to be able to intercommunicate -- but the only requirement is that they speak some standard language in the world – but they are not required to speak the same language – even in the same department, - and no translators are required either.   They will not all be able to call each other.

      (In the TEITAC requirements, **each system <u>was</u> allowed choose its own "language"** (its own RTT format) **<u>but</u> within each system everything had to speak that chosen "language" (**RTT format**)**; **- AND - where a system connected to another, the gateway had to translate from the 'one' language of first system to the 'one' language of the second system*.***
      ii. So in the TEITAC consensus report each system (IMS, SIP, XMPP, SKYPE, etc) **<u>could select their own RTT format</u>** – but everything within that system needed to support that format – and the systems needed to translate where they joined with each other). This is what is required by industry for all of its voice systems - but requiring the same thing for Real-time text was fought and removed.

4. **…is written to only apply to "user equipment" meaning that network equipment and systems installed in agencies etc have no requirement to support RTT.**
   a. Thus there is no mechanism for the RTT to actually work in real world systems that include switches, gateways, and other call processing equipment between the phones or terminal calling devices.

How to fix them

A longer discussion of the above points is available on request. However it is more useful to talk about how to fix the provisions. Red-lining to repair the current language is provided below - -with annotations. Only critical corrections are noted. Only 3 provisions need to be repaired. But each word and note is important if the edits are not to be undone again. Rationales are provided for each.

6.2 Real-time text (RTT) functionality

6.2.1 RTT provision

6.2.1.1 RTT communication

Where ICT supports ~~voice~~ VoIP communication~~ in a specified context of use,~~ and its software runs on hardware that has the ability to display multiline text and generate text, ~~in a specified context of use~~, the ICT shall ~~allow~~ enable a user to communicate with another user by RTT and voice concurrently on the same call session.

~~NOTE 1: The RTT capability can be provided as a factory default or added later.~~

NOTE 1: If the platform provides RTT functionality, the ICT can make use of it but needs to integrate it naturally within the ICT in the same way a print routine appears to be a natural part of a program even though integrated from the platform.

~~NOTE 2: Provision of the RTT capability may require additional service provision, additional hardware and/or software which may be provided separately or together..~~

NOTE 2: The availability of voice and RTT running concurrently can allow the RTT to replace or support voice and transfer additional information such as numbers, currency amounts and spelling of names.

NOTE 3: This provision does not require that a keyboard or display be added to a product just to support RTT.

RATIONALE
• "VoIP" was added because these provisions should be applied only to VoIP and not PSTN. • ~~in a specified context of use~~ was removed since nowhere is this defined, and having a standard or regulation that state that it must be met but doesn't specify things until later is untestable. Also the mfgr can't design a product this year to meet something that won't be specified til later. • "and is on hardware that has the ability to display multiline text and generate text" was added because these provisions should **NOT** be required for those ICT that cannot already generate or display text for other reasons. • ~~"at least one element~~" was removed since the RTT should be part of the ICT not some other ICT on the device. • "on the same call session" was added – per discussion above – since the elder who cannot hear well and cannot type – needs to speak and see text back on the same call. • Old Note 1 was removed – since adding it later means no requirement, does not integrate with app, and creates interoperability problems with the ICT • Old note 2 was removed because it serves no purpose with the added text above exempting ICT that cannot generate or display text. RTT itself does not require new hardware. • The phrase ~~which may be provided separately or together~~ was removed since it basically states that the RTT can be provided by a third party as a completely separate item from the device that does the voice (e.g. a phone complies if it can be used for voice and a computer can be use for text. • New Note 1 added to ensure that this is allowed and understood • New Note 2 is just moved up from old 6.3.1.2 • New Note 3 was added to ensure the provision could not be misread as requiring the addition of a keyboard or display if not already part of the product design.

6.2.1.2 Concurrent voice and text

Where a network component ~~ICT~~ supports two-way voice communication in a specified context of use, ~~and enables a user to communicate with another user by RTT,~~ it shall provide a mechanism to ~~select a mode of operation~~ allow~~ing~~ concurrent voice and text.

~~NOTE: The availability of voice and RTT running concurrently can allow the RTT to replace or support voice and transfer additional information such as numbers, currency amounts and spelling of names.~~

RATIONALE - The old 6.3.1.2 provision is now redundant with #1 above – so it was changed to cover requirement on systems, which was missing.

6.2.2 Display of Real Time Text

6.2.2.1 Visually distinguishable display

Where ICT has RTT send and receive capabilities, displayed sent text shall be visually differentiated from and separated from received text.

6.2.2.2 Programmatically determinable send and receive direction

Where ICT has RTT send and receive capabilities, the send/receive direction of transmitted text shall be programmatically determinable, unless the RTT has closed functionality.

NOTE: The intent of clause 6.3.2.2 is to enable screen readers to be able to distinguish between incoming text and outgoing text when used with RTT functionality.

6.2.3 Interoperability

Where ICT with ~~RTT~~ two way voice communication functionality, interoperates with other ICT networks ~~with RTT functionality (as required by 6.2.1.1), they~~ it shall support at least ~~one of the four~~ the RTT interoperability mechanism~~s~~ established for that system as described below:

a) Where ICT interoperates ~~ing~~ over the Public Switched Telephone Network (PSTN), ~~with other ICT that directly connects to the PSTN as described in~~ it shall support Recommendation ITU-T V.18 [i.22] or any of its annexes for text telephony signals at the PSTN interface;

b) Where ICT interoperates ~~ing~~ with other ICT using VOIP with Session Initiation Protocol (SIP) it shall support ~~and using~~ real time text that conforms to RFC 4103 [i.13]

c) Where ICT interoperates ~~ing~~ with other ICT using ~~RTT that conforms with~~ the IP Multimedia Sub-System (IMS) set of protocols specified in TS 126 114 [i.10], TS 122 173 [i.11] and TS 134 229 [i.12] it shall support the RTT that is specified for those protocols;

d) Where ICT interoperates ~~ing~~ on any other systems, it shall support the ~~with other ICT using a~~ published and available common specification for RTT exchange that is published and available and specified as the common Real-time text Interoperability format for that system. ~~This common~~ That specification shall include a method for indicating loss or corruption of characters.

e) Where ICT systems that support voice communication connect to other systems, the gateway between the two systems shall support transcoding to and from the established RTT formats of the two systems.

NOTE: Any ICT can support additional RTT formats (that they would use when they encounter a second device that also supports the alternate format), as long as the ICT first supports the named RTT format for the system it is operating on so that a successful connection is always ensured.

RATIONALE

- This was changed slightly (but critically) to require that each system have a format for RTT that everything on the system supports, and that is transcoded at the borders with other systems. This is the only way that devices on systems can interoperate.
- ~~that provide RTT functionality~~ was replaced with two way voice communication functionality since qualifying this with "RTT functionality" can create a chicken and egg situation where terminals are not required to support RTT till networks do – and networks won't if terminals don't. Terminals should support the RTT format decided by the network. And Network components should too.
- For ***those systems where industry has already created a standard format,*** that format is named. For all others, ***each system can name its own format*** – as long as everything on that system supports that single format
- The NOTE was added to make it clear that companies are not limited to the common interoperability format – and can support other and new formats – as long as they also support the interoperability standard.

6.2.4 Real-time text responsiveness

RTT input shall be transmitted to the ICT network supporting RTT within 1 second of the input entry.

NOTE 1: Input entry is considered to have occurred when sufficient user input has occurred for the ICT to establish which character(s) to send.

NOTE 2: Input entry will differ between systems where text is entered on a word-by-word basis (e.g. speech-to-text and predictive-text based systems) and systems where each character is separately generated.

CLEAN COPY of the EDITS TO M376 Above

6.2 Real-time text (RTT) functionality

6.2.1 RTT provision

6.2.1.1 RTT communication

Where ICT supports VoIP communication and its software runs on hardware that has the ability to display multiline text and generate text, the ICT shall enable a user to communicate with another user by RTT and voice concurrently on the same call session.

NOTE 1: If the platform provides RTT functionality, the ICT can make use of it but needs to integrate it naturally within the ICT in the same way a print routine appears to be a natural part of a program even though integrated from the platform.

NOTE 2: The availability of voice and RTT running concurrently can allow the RTT to replace or support voice and transfer additional information such as numbers, currency amounts and spelling of names.

NOTE 3: This provision does not require that a keyboard or display be added to a product just to support RTT.

6.2.1.2 Concurrent voice and text

Where a network component supports two-way voice communication in a specified context of use, it shall provide a mechanism to allow concurrent voice and text.

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NOTE: The intent of clause 6.3.2.2 is to enable screen readers to be able to distinguish between incoming text and outgoing text when used with RTT functionality.

6.2.3 Interoperability

Where ICT with two way voice communication functionality, interoperates with other ICT networks it shall support at least the RTT interoperability mechanism established for that system as described below:

f) Where ICT interoperates over the Public Switched Telephone Network (PSTN), it shall support Recommendation ITU-T V.18 [i.22] or any of its annexes for text telephony signals at the PSTN interface;

g) Where ICT interoperates with other ICT using VOIP with Session Initiation Protocol (SIP) it shall support real time text that conforms to RFC 4103 [i.13]

h) Where ICT interoperates with other ICT using the IP Multimedia Sub-System (IMS) set of protocols specified in TS 126 114 [i.10], TS 122 173 [i.11] and TS 134 229 [i.12] it shall support the RTT that is specified for those protocols;

i) Where ICT interoperates on any other systems, it shall support the published and available common specification for RTT exchange that is published and available and specified as the common Real-time text Interoperability format for that system. That specification shall include a method for indicating loss or corruption of characters.

j) Where ICT systems that support voice communication connect to other systems, the gateway between the two systems shall support transcoding to and from the established RTT formats of the two systems.

NOTE: Any ICT can support additional RTT formats (that they would use when they encounter a second device that also supports the alternate format), as long as the ICT first supports the named RTT format for the system it is operating on so that a successful connection is always ensured.

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NOTE 2: Input entry will differ between systems where text is entered on a word-by-word basis (e.g. speech-to-text and predictive-text based systems) and systems where each character is separately generated.

Alternately – Use Access Board ANRPM language

Here is the language of the Access Board ANPRM – revised to provide similar functionality and avoid the above issues.

408 ICT with Two Way Voice Communication

408.1 General. ICT that provides two-way voice communication shall conform to 408.

<SNIP>

408.6 Real Time Text Functionality. Where ICT provides real time voice communication, ICT shall provide simultaneous real time text functionality on the same call session and shall conform to 408.6.

> **RATIONALE:** See above regarding the need for voice and text to be on the same call. Same call session is critical or it won't work for elders who cannot hear or hear well and also cannot type (vision or too slow or arthritis and cannot type at all so must speak and receive text back).
>
> **NOTE – this was in earlier version and we think it was inadvertantly dropped.**

408.6.1 Display of Real Time Text. Where ~~provided,~~ ICT has, or is software that runs on a platform that has, a multi-line displays it shall ~~be compatible with~~ support real time text compatible with the real-time text system~~s~~ used on the network.

> **RATIONALE:** Old language said "displays must be compatible with text". This is always true of displays, but the wording didn't actually require the ICT to be compatible. The provision needs to be "ICT WITH displays shall SUPPORT RTT".
>
> We THINK this is what the old language was meant to say – but didn't. So this would be editorial cleanup?

408.6.2 Text Generation. Where ~~provided,~~ ICT has, or is software that runs on a platforms that has, features capable of text generation it shall ~~be compatible with~~ support real time text compatible with the real-time text system~~s~~ used on the network.

> **RATIONALE:** Same problem as above. As originally worded it said 'where features are provided that generate text – they shall support RTT'. Features that generate text (e.g. keyboards) always work for RTT. The requirement needs to be that the **ICT** will support text – not just the "keyboard". Also just a cleanup mistake?

408.6.3 Interoperability. Where ICT interoperates outside of ~~its~~ a closed system, or where ICT connects to other systems, ICT shall conform to 408.6.3.1 or 408.6.3.2 or 408.6.3.3.

408.6.3.1 PSTN. Where ICT interoperates with the PSTN (Public Switched Telephone Network), real time text shall conform to the TIA 825-A (incorporated by reference in Chapter 1) Baudot standard for TTY signals at the PSTN interface.

408.6.3.2 VoIP Using SIP. Where ICT interoperates with Voice over Internet Protocol (VoIP) products or systems using Session Initiation Protocol (SIP), they shall support transmission of real time text that conforms to RFC 4103 (incorporated by reference in Chapter 1).

408.6.3.3 Where ICT interoperates on any other systems open or closed, it shall support the common specification for RTT exchange that is specified for that system by the body responsible for creation of specifications for that system.

RATIONALE: Need to have RTT on all other systems too – but need to allow industry to choose the RTT for each system.   We think this flexibility for industry is important.

408.6.4 Real Time Text Compatibility. Voice mail, auto-attendant, and interactive voice response telecommunications systems shall be compatible with real time text that conforms to 408.6.3.

Original TEITAC Language

1-C: Pass Through

Products that transmit or conduct information or communication must preserve accessibility information that is transmitted in non-proprietary, industry-standard codes, translation protocols, or formats.

Technologies that use encoding, signal compression, format transformation, or similar techniques must not remove information needed for access, or must restore it upon delivery.

Firewalls, routers, gateways and other products that pass real-time voice communication must also pass REAL-TIME TEXT communication signals (including mixed voice and REAL-TIME TEXT) that are standard in the United States for that technology platform without distortion or error beyond 1%.6. Additional Requirements for Real-Time Voice Conversation Functionality

6-A: Real-Time Text Reliability and Interoperability

If hardware or software provides real-time voice conversation functionality it must provide at least one means of REAL-TIME TEXT communication where the following reliability requirements are met:

1. Products must use a REAL-TIME TEXT (RTT) system that meets the following requirements:

   1. RTT format must be a standard REAL-TIME TEXT format for the voice platform that is supported by all TERMINAL, router, gateway and other products on that platform;
   2. RTT format must transmit characters with less than 1 second delay from entry;
   3. RTT system must transmit TEXT with less than 1% Total Character Error Rate at the peak network traffic specified for intelligible speech transmission (TEXT must work on the network as long as speech does);
   4. The RTT system, together with the audio system, must support speech and TEXT in both directions in the same call session (and support speech and TEXT simultaneously in both directions in the same call session if IP based)
   5. RTT system must not utilize audio tones for transmission of REAL-TIME TEXT over IP. Note: this is subject to a waiver of the TTY support requirement from the FCC for systems that implement IP based RTT. Also subject to consumer acceptance of prefixes or phone numbers to direct TTY traffic to gateways capable of handling TTY translation.

2. Where products or systems interoperate outside of their closed systems, they must:

   a. If product interfaces with PSTN, it must use TIA 825A Baudot where it interfaces to the PSTN.
   b. If product interfaces with other VoIP products or systems (outside of a self-contained product-system) using SIP it must support transmission of TEXT as per XXX where it interfaces with other VoIP products or systems. Note: this is subject to a waiver of the TTY support requirement from the FCC for systems that implement IP based RTT. Also subject to consumer acceptance of prefixes or phone numbers to direct TTY traffic to gateways capable of handling TTY translation.
   c. If product connects to other products or systems using a protocol other than SIP it must use the standard REAL-TIME TEXT protocol that meets provision 1 above that has been established for that protocol.

   Note 1: RFC-4103, TIA 1001, and MSRP (RFC4975) are being explored to fill the role of XXX. The intention is that XXX will be replaced by one interconnection format in all places it was used.

   Note 2: All products may support and use other protocols in addition to these as long as they meet the 5 requirements of 5-B(1) above.

   Note 3: A self-contained SIP system that uses the same real-time text protocol can be treated as a single product and can use any protocol internally as long as it supports XXX where the system-product connects to other systems or products.

Rationale: This provision, along with 6-B, allows people with disabilities to communicate using standard IP methods rather than continuing to support TTY within IP networks and devices.

6-B: Voice Terminal Hardware and Software

TERMINAL hardware or software that is capable of providing voice communications in real-time must comply with the following:

1. Receive only: If hardware or software TERMINAL provides voice conversation over IP in any form, and has a user interface with a multi-line display or a user interface that runs on devices that have a multi-line display, then that TERMINAL must display any REAL-TIME TEXT that is received if it is received in the format for the voice and REAL-TIME TEXT system being used on the network on which it is installed.
2. Send and Receive: If TERMINAL hardware or software provides voice conversation over IP in any form, and has TEXT generation capability, then the TERMINAL must allow users to send REAL-TIME TEXT in the format for the voice and REAL-TIME TEXT system being used on the network on which it is installed.
3. If IP TERMINAL hardware or software does not provide REAL-TIME TEXT send and receive capability then the TERMINAL must support the addition of TERMINALS and TERMINAL peripheral equipment that support REAL-TIME TEXT functionality in conjunction with the voice call functionality, in the same location and with the same permissions for use as their voice TERMINAL. If the TERMINAL is in a public or shared area and not in an individual's private work area then the connection must be possible [without requiring system-administrator intervention]. Note: the "without system-administrator intervention" is a serious concern due to security issues, but removal would prevent people from connecting devices outside of their home system. Additional work is needed to address this issue.]
4. If TERMINAL is analog or TDM-digital wired TERMINAL then it must support the connection of a TTY via an RJ-11 jack in the same location and with the same permissions for use as the telephone and it must be capable of allowing simultaneous speech and TEXT conversation without interference or its microphone must be capable of being turned on and off to allow the user to intermix speech with text use.

Note 1: Provision of the RJ-11 jack may be accomplished through one of the following techniques:

a. provision of the RJ-11 jack on the telephone,
b. the use of a Y-adapter that allows both the analog telephone and the TTY to be plugged into the same line outlet,
c. having built in capability to support an RJ11 module that can provide a connection point for TTYs.

Note 2: The standard format for PSTN is TIA-825A. For SIP is it XXX. For other voice transport protocols the format is to be determined by the entity responsible for the voice transport protocol.

6-C: IVR, Auto-Attendant and Messaging

Voice mail, messaging, auto-attendant, and interactive voice response TELECOMMUNICATIONS SYSTEMS must provide access in the following manner:

1. All functions that are accessible to voice users must also be directly accessible to users of REAL-TIME TEXT.
2. Use the ITU-T G.711 recommendation <SNIP>